An Examination of the Effects of Spectral Cues and Pitch Anomalies on Melody Recognition in Cochlear Implants

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Abstract

As cochlear implant patients achieve greater success in understanding speech it is important to examining more complex hearing tasks. One logical step is to begin exploring some of the problems associated with music perception. Because music requires relatively precise pitch information, it can present unique challenges to current speech-oriented implant designs, which are limited in the amount of spectral information they can present. In addition, anomalies associated with channel interactions that have been shown to effect speech perception may have even greater effects on music perception.

Two pilot studies were performed to examine both the effects of spectral cues and pitch anomalies on melody recognition. Both studies were done on normal hearing patients using acoustic models. For the first study, melody tokens were processed using two types of spectral models: all spectral channels presented for each time sample (ranging from 2 to 32 channels) and 6 of 20 spectral channels presented for each time sample. The resulting melody tokens were used in a recognition test in which subjects were presented with each token and asked to identify the token from a closed set. The results suggest that there is a positive correlation between the number of spectral channels and melody recognition. For the second study the processing model was altered to account for pitch reversals and missing channels. These two types of pitch anomalies have been shown to have detrimental effects on speech perception. Once again the processed tokens were used in a recognition test in which subjects were asked to identify the processed melody tokens from a closed set. The results show that these types of pitch anomalies do not affect melody recognition as significantly as they have been shown to affect speech perception.

In order to examine the effects of spectral cues and channel interactions in more detail, studies on cochlear implant patients must be done. Such studies require an interface that allows the researcher to directly control the stimulation patterns of implanted electrodes. As a result, stimulation programs are being designed for use on a Clarion® Research Interface. A completed stimulation program would facilitate the testing of cochlear implant patients and allow this study to be expanded.

I. Introduction

The cochlea is a snail shaped fluid filled organ in the inner ear. It is responsible for translating mechanical vibrations created by acoustical waves into nerve firings that are interpreted by the brain as sound. Acoustical energy passing through the outer and middle ear causes the flexible basal membrane within the cochlea to vibrate. The vibrations displace hair cells attached to the membrane. The displaced hair cells then release electrochemical substances that are sensed by adjacent auditory neurons (Better Hearing, 2005).

The most common form of hearing loss is sensorineural hearing loss. It is estimated that as many as 17 million Americans experience some degree of sensorinerual hearing loss (American Speech, 2005). This type of hearing loss occurs as a result of damage to auditory hair cells or auditory neurons. Significant damage can cause severe or profound deafness. Studies have shown however, that the most common cause of deafness is the loss of hair cells rather than auditory neurons. Thus many people with sensorineural hearing loss still possess the neural hardware necessary to interpret sound. Cochlear implants are able to bridge the gap between acoustical energy and neural stimulation by using an array of electrodes to electrically stimulate the auditory neurons. The electrode array works in conjunction with a signal-processor that converts acoustic waves into electric signals and a transmission system that passes information from the external signal processor to the internal electrodes. Modern cochlear implants differ primarily in electrode array design and signal processing scheme.

There are several strategies for designing electrode arrays. The designs differ in placement, number, spacing and configuration of the individual electrodes. Placement refers to the insertion depth as well as the implants position in or near the cochlea. One effective placement technique is to insert the electrodes into the scala tympani. This allows the electrodes to more closely emulate the natural spatial relationship between stimulation position and frequency. Similar to a normal cochlea, this placement allows higher frequencies to stimulate neurons near the base while lower frequencies stimulate neurons near the apex (Loizou, 1998).

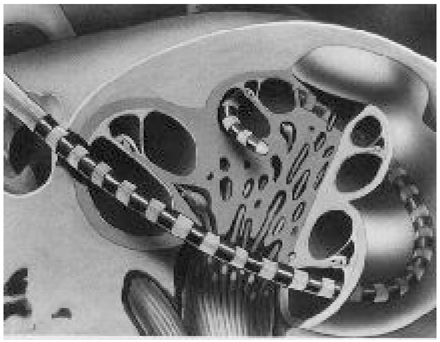


Figure 1: Cross Section of implanted cochlea. The electrode array enters the scala tympani near the basal end at the bottom right (Fearn, 2004).

Another design consideration is the number of electrodes. Having multiple electrodes makes it possible to exploit the spatial/frequency relationship. In general, an increase in the number of electrodes leads to finer place resolution for coding frequencies and the ability to present more spectral information. However, because electrical stimulation tends to create pulses that propagate symmetrically from the activation point there is an upper limit to the number of electrodes that can be effective. As the number of electrodes increases, the decrease in spacing leads to interference which can reduce the effectiveness of the entire array (Loizou, 1998).

Interferences that occur between electrodes are known as channel interactions. There are several types of interactions that have been shown to have detrimental effects on speech perception in cochlear implant patients. They include indiscriminable electrodes, forward masking effects, and pitch reversals. Electrodes are deemed indiscriminable when two or more frequency bands are identically perceived. Forward masking occurs when a prior stimulation pattern causes a stimulus to be imperceptible. Pitch reversals describe the non-monotonic changes in pitch perception that can occur along the cochlea from apex to base. (Throckmorton and Collins, 2002).

Before an input signal can be presented to the implanted electrodes, it must be processed so that it can be effectively mapped to the electrodes. The signal-processing unit is responsible for this mapping. This is the most complex component of the cochlear implant. Not surprisingly, this is also where the most variation in design can be found. Despite the variety, most processing schemes can be described as either waveform extraction or spectral feature extraction schemes.

Both the Compressed-Analog (CA) and Continuous Interleaved Sampling (CIS) approaches focus on persevering wave information. Both techniques divide the signal into frequency bands and send processed forms of these frequency bands to the electrodes. The difference between the two designs lies in the way in which each approach delivers the signal to the electrodes. The CA approach sends the waveforms simultaneously to all four of its electrodes while the CIS uses non-simultaneous pulses to deliver the waveform (Loizou, 1998).

One of the Nucleus designs uses formant extraction algorithms to present spectral information to the electrodes. The original design extracted only a single formant. Subsequent innovations led to models that extracted the fundamental frequency and the second formant (F0/F2) and eventually to a processor like the Miniature Speech Processor (MSP), which used 3 formats (F0/F1/F2) and extracted high frequency information (Loizou, 1998).

Though very different, all three of these techniques have proven effective for speech perception in commercial designs (Boex *et. al.,* 1994; Mercenich *et al.,* 1984; Loizou, 1998). As speech perception continues to improve, one logical next step is to examine more complex hearing tasks such as music perception. Music is an important part of the daily lives of millions. Unfortunately many implant patients report that music sounds unnatural and is often unrecognizable. In order to understand why current speech processing techniques are inadequate for musical perceptions, it is necessary to examine how a normal human ear perceives music.

Musical perception is based on rhythm, timbre, and pitch. Rhythm is related to the temporal characteristics of the sound. Timbre is more difficult to describe but can be defined as ``the quality of a sound by which a listener can tell that two sounds of the same loudness and pitch are dissimilar'' (Fragoulis, 1999). Pitch conveys melody and is strongly related to the spectral content of the sound. For accurate musical perception, all three of these characteristics must be transmitted. Speech perception requires accurate transmission of temporal information. In addition, understanding vowels and distinguishing between voiced and unvoiced sounds requires perception of information related to timbre. Therefore successful speech processing designs must and in fact do transmit both rhythm and timbre relatively well. However, in non-tonal languages such as English, pitch conveys less important information such as the sex of the speaker and whether a sentence is a question or a statement. Therefore, less emphasis has been placed on creating implants that can adequately transmit pitch (Fearn, 2001). Pitch perception turns out to be important not only for musical perception, but also for intonation, speaker identification (Wouters, 2003), and in tonal languages such as Mandarin, Cantonese and Thai distinct ideas can be conveyed through pitch (Nie, 1998).

Because music requires relatively precise pitch information, it can present unique challenges to current speech-oriented implant designs, which are limited in the amount of spectral information they can present. In addition, anomalies associated with channel interactions that have been shown to effect speech perception may have even greater effects on music perception. As a result, two pilot studies were performed to examine both the effects of spectral cues and pitch anomalies on melody recognition. Both studies were done on normal hearing patients using acoustic models. This work is an extension of work done my Kong *et al.* in which he investigated the effect of limited spectral information by measuring the ability of subjects to identify melodies. He found that as many as 32 channels may be needed for effective melody recognition.

II. Methods

The experiment has progressed through three major stages. The first objective was to become familiar with current speech processing techniques by designing a functional speech-processing model in Matlab. Though such models already exist, creating one was important for understanding signal processing techniques and cochlear implant acoustic models. Next, acoustic models were used to perform pilot studies on normal hearing patients. The objective of these studies was to assess the effects of spectral cues and pitch anomalies on melody recognition. Finally, design was begun on a research interface that would implement a frequency discrimination test for cochlear implant subjects.

A. Speech Processing Model

A functional signal-processing model was designed using Matlab. The model implements the 8F and 6/20F algorithms. Both algorithms are outlined extensively by Throckmorton (2001). The 8F and 6/20F models Throckmorton describes are similar to the CIS (Continuous Interleaved Sampling) and SPEAK (Spectral Peak) strategies respectively. Both algorithms pre-filter the input waveform using a 1200Hz second order high-pass butterworth filter. For the 8F algorithm, the signal is then divided into eight frequency bands. The cutoff

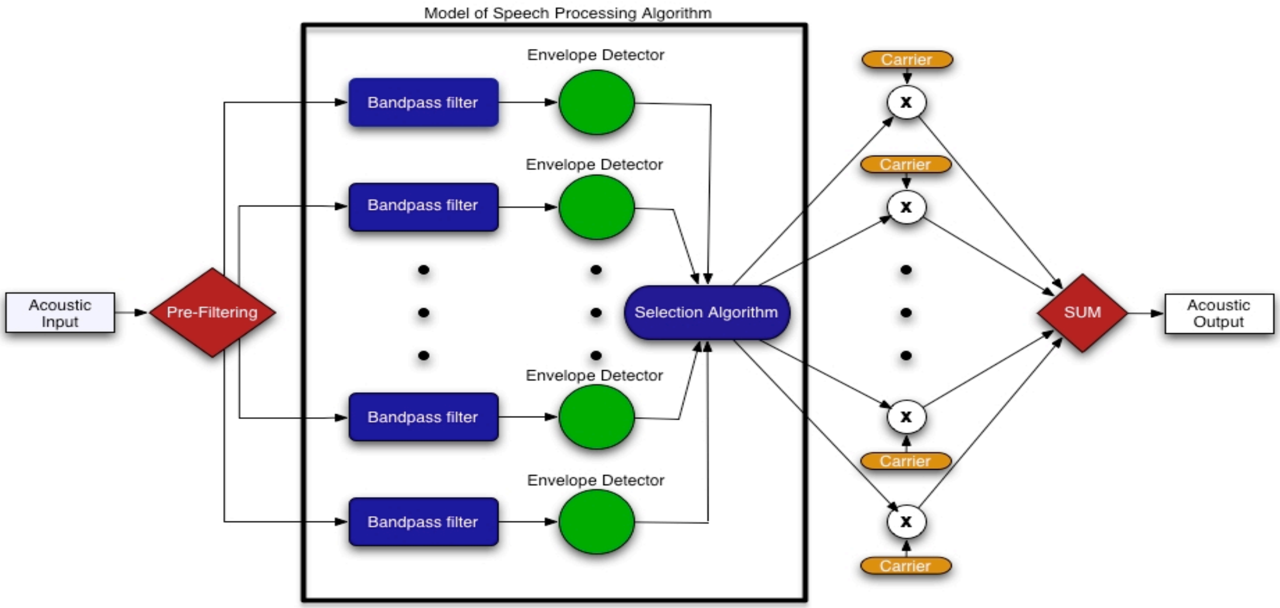


Figure 1: Diagram of Speech Processing Model

frequencies of the sixth order butterworth filters are logarithmically spaced between 150Hz and 6450Hz. For the 6/20 algorithm, the signal is divided into twenty frequency bands. The cutoff frequencies for the first seven bands are approximately linearly spaced between 250Hz and 1550Hz. The remaining thirteen bands are logarithmically spaced between 1550Hz and 10823Hz. Logarithmic spacing was used to approximate equal cochlear distance as defined by the Greenwood map (Greenwood 1990).

Next, the resulting signals (8 signals for 8F algorithm and 20 for 6/20 algorithm) are envelope detected by taking their absolute values and filtering them with a 200Hz low pass butterworth filter. For the 6/20F strategy, reconstruction of the signal required the use of a selection algorithm to reduce the number of envelopes from 20 to 6. To accomplish this, the signals were broken into 2ms windows. Within each time window, the 6 envelopes of greatest magnitude were selected. For the 8F strategy no such selection algorithm was required. For both schemes, the resulting envelopes were multiplied by a sine wave with a frequency equal to the center frequency of the bandpass filter that produced it. The modulated signals were then summed and the result was outputted to the *wavwrite* function of Matlab.

B. Melody Recognition

1. Stimuli

Sound files for twelve familiar melodies were created using ReBirth RB 338 version 2.0.1. Each of the twelve melodies was recorded with and without rhythm. Specific information about the melodies is included in the Figure A1 of the Appendix. The songs are the same as those used in a melody identification study perform by Ying-Yee Kong *et al*. (2004). Kong *et al*. (2004) performed recognition studies on a group of normal-hearing patients in order to select songs that would be easily recognizable.

2. Processing Models

a. Spectral Models

The stimuli were processed using a modified version of a cochlear implant simulation program designed by Chandra S. Throckmorton. Each melody was profiteered using a first order high-pass Butterworth filter with a cutoff frequency of 1kHz. In addition a sixth-order low-pass Butterworth filter was used to remove frequencies greater than half the sampling frequency. This was done to prevent aliasing.

The prefiltered melodies were processed using five different algorithms. All of the processing methods used sixth-order Chebyshev type I bandpass filters. Two of the algorithms were designed to be similar to the CIS (e,g.Kessler, 1999; Wilson *et al*., 1991) and SPEAK (e.g., Whitford *et al*., 1995) processing strategies. The CIS-type algorithm (8/8F) used a bank of 8 bandpass filters with center frequencies logarithmically spaced between 150 and 6450 Hz. The SPEAK-type algorithm (6/20F) used a bank of 20 bandpass filters with center frequencies spaced linearly between 250 and 1600 Hz and logarithmically between 1600 and 10823 Hz. The 8/8F algorithm presented acoustical information from all eight filters in each window while the 6/20F presented information only from the six filters with greatest energy. The three remaining algorithms used banks of 32, 8, or, 2 bandpass filters (32F, 8F, 2F). For each of these algorithms, the center frequencies were logarithmically spaced between 80 and 8800 Hz. Approximately equal cochlear distance was used for each band and the cutoff frequencies were determined using the Greenwood Map (Greenwood, 1990).

Envelopes of the bandpass filters were extracted every 2ms by full-wave rectifying the bands and low-pass filtering them using an eighth-order Chebyshev type I filter. Once again, assuming approximately equal cochlear distance, sinusoidal carrier signal frequencies were calculated based on the cutoff frequencies of each bandpass filter using the Greenwood function. The carrier signals were amplitude modulated by the root-mean-square energy of the envelopes. The output melody was created by summing the modulated carrier signals.

b. Pitch Reversal Model

In order to analyze the effects of channel interactions on musical perception, the Throckmorton model was further altered to account for pitch anomalies. In this pilot study, anomalies were only taken into account for the 6/20F algorithm. Three types of anomalies were tested (low-frequency, mid-frequency, and high-frequency) using two different methods. The first model reassigned carrier signals affected by pitch reversals. Psychophysical studies on cochlear implant subjects provide no information regarding the exact frequency to which these carrier signals should be reassigned (Throckmorton and Collins 2002). Therefore, the frequencies for this experiment were based on the reassignments made by Throckmorton and Collins (2002) which put the reversal frequencies equidistant between unaltered frequencies to maximize discrimination between frequencies and minimize changes to the pitch range. The three resulting models for this method were described as PRL (pitch reversal low-frequency), PRM (pitch reversal mid-frequency), and PRH. The carrier frequencies for the pitch reversal model are listed in Figure A2 of the Appendix. For the second model, carrier signals affected by pitch reversals were discarded rather than reassigned. The amplitudes of affected signals were set to zero thus reducing the total number of effective filters from 20 to 15. Six tones are still selected and presented in each window however. The resulting models for this method were described as PGL (pitch gap low-frequency), PGM, and PGH.

3. Experiment I: Spectral Cues

For this pilot study, six normal hearing listeners were selected from the research group. More information about the subjects is included in Figure A3 of the Appendix. Before testing began, there was a training section in which the subjects were encouraged to listen to each of the original melodies with and without rhythm. For testing, the melodies were presented in 12 groups, one for each condition. The order of the groups was the same for all listeners (Original/Rhythm, Original/No Rhythm, 32F/Rhythm, 32F/No Rhythm, 8F/Rhythm, 8F/No Rhythm, 2F/Rhythm, 2F/No Rhythm, 6/20F/Rhythm, 6/20F/No Rhythm, 8/8F/Rhythm, 8/8F/No Rhythm). Within the groups, each melody was presented three times in random order. The names of all twelve melodies were shown on the computer screen and the listeners were asked to identify the correct melody from this closed set. Repetition was not allowed and visual feedback was given immediately after each response.

4. Experiment II: Pitch Reversals

In a second experiment, 5 of the previous 6 subjects were retested. Subject 3 was not available. There was no practice section for this test. Only the 6/20F processing algorithm was used, but all six of the pitch reversal models were tested. The order of the groups was fixed (Original/Rhythm, Original/No Rhythm, PGH/Rhythm, PGH/No Rhythm, PGL/Rhythm, PGL/No Rhythm, PGM/Rhythm, PGM/No Rhythm, PRH/Rhythm, PRH/No Rhythm, PRL/Rhythm, PRL/No Rhythm, PRM/Rhythm, PRM/No Rhythm). Within the groups each melody was presented twice in random order. The visual interface was very similar to Experiment I. As before, listeners were asked to identify the correct melody from a closed set, repetition was not allowed, and visual feedback was given immediately after each response.

C. Frequency Discrimination Model

1. The CLARION® Research Interface

The CLARION® Research Interface (CRI) created by Advanced Bionics Corporation consists of several hardware components: a host PC, a Digital Signal Processing (DSP) Development Board, a CLARION® Speech Processor headpiece, and an Implantable Cochlear Stimulator (ICS). A Motorola DSP56002PV80 was used as the DSP chip. Stimulus applications could be loaded to the DSP board and were created using utility programs installed on the host PC. Outputs from the ICS were examined using a Tektronix TDS 210 Two Channel Digital Real Time Oscilloscope.

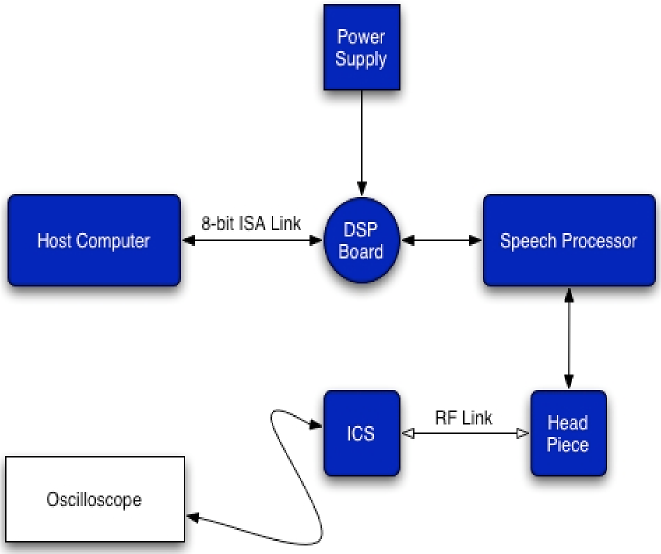


Figure 3: Diagram of CLARION® Hardware Components

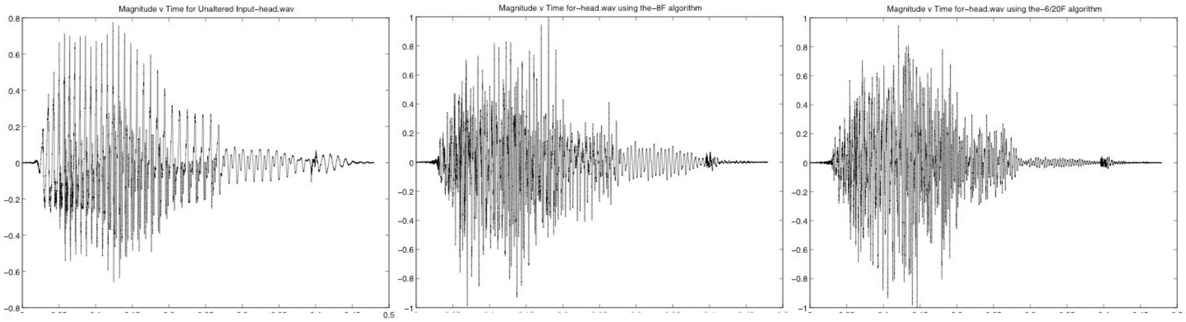
2. Creating the Model

The goal of this stage of the project was to utilize the Clarion® Research Interface to create psychophysical tests for examining frequency discrimination limens in cochlear implant patients. Pfingst *et. al* (1994) outlines a method for finding discrimination limens as a function of reference stimulus level in patients with Nucleus-22 cochlear implants. Though similar to the Pfingst *et al.* design, a model for this experiment would be based on 8 rather than 22 electrode hardware. Though strides have been made in understanding the interface, a functional model has not yet been completed.

III. Results

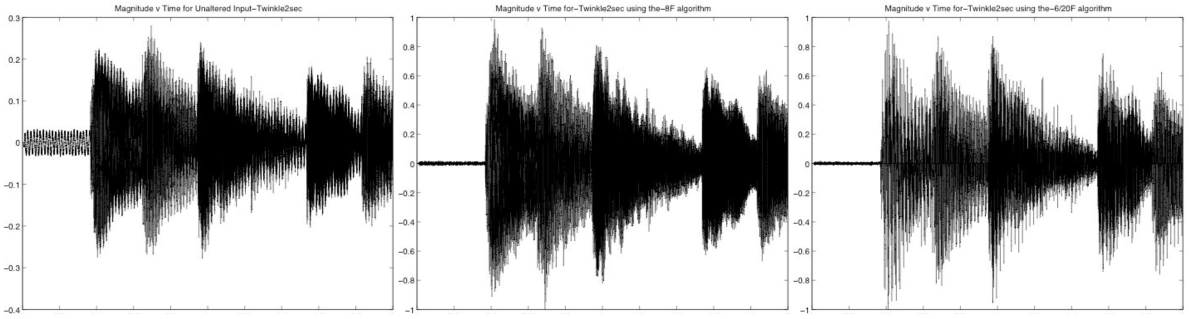
A. Speech Processing Model

The speech-processing model implements both algorithms effectively. Although this model was based on current speech processing designs, it was tested with both speech and music tokens. The output plots show that the processing techniques preserve the basic characteristics of the input signals. Acoustically the output signals from both the 8F and 6/20F algorithms are recognizable for speech and basic music tokens. There is notably distortion in the output signals, but given the nature of the processing some distortion is expected.



(A) Unprocessed Speech Token **(B) Processed with 8F algorithm** **(C) Processed with 6/20 algorithm**

Figure 4: Speech Token Waveforms for "Head"



(A) Unprocessed Music Token **(B) Processed with 8F algorithm** **(C) Processed with 6/20F algorithm**

Figure 5: Music Token Waveforms for "Twinkle Twinkle Little Star"

B. Melody Recognition

1. Experiments I: Spectral Cues

For original melodies and melodies with 32 bands of frequency content, there was no significant difference between recognition scores for the rhythm and no rhythm conditions. However for the 6/20F, 8/8F, 8F, and 2F melodies the differences between recognition scores for rhythm versus no rhythm were increasingly significant with p-values of 0.031, 0.042, 0.007, and 0.004 respectively.

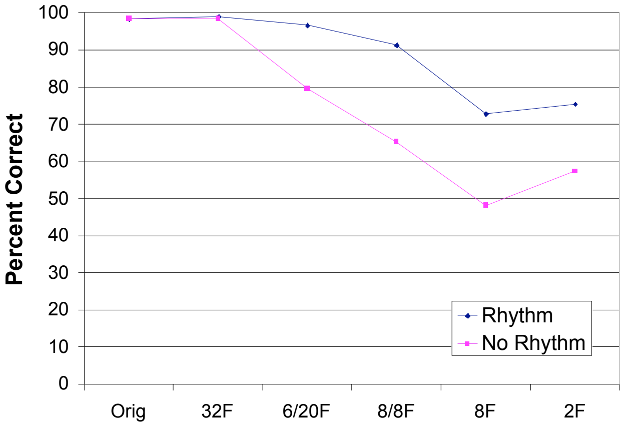


Figure 6: Melody recognition scores for spectral models as a function of processing algorithm

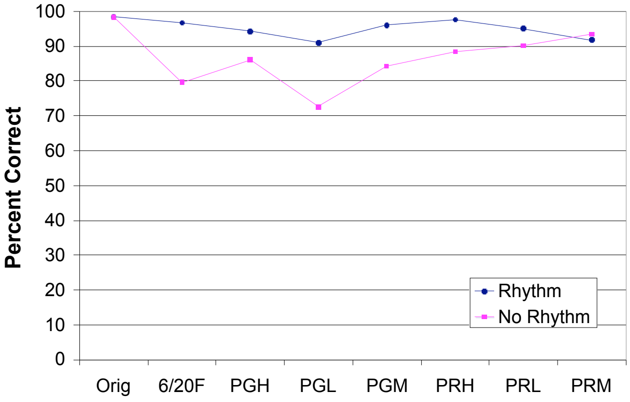


Figure 7: Melody recognition scores for pitch reversal models as a function of processing algorithm

2. Experiment II: Pitch Reversals

Because the unaltered 6/20F model was not tested in this experiment, information from Experiment I was used as a basis for comparison and is included in Figure 2 for 6/20F. The remaining data including that for original melodies was obtained from Experiment II. When compared to the unaltered model (from Experiment I), the pitch reversal models had no significant effect on melody perception for both the rhythm and no rhythm conditions. However the PGL scores for both rhythm and no rhythm conditions were significantly different from scores for the original melodies with p-vales of 0.038 and 3.64E-5 respectively.

C. Research Interface

After extensive research progress was made toward creating a successful model. Methods for controlling frequency, amplitude, pulse type, and pulse duration were created. Using the oscilloscope changes to these parameters could be evaluated and measured. However, several problems prevented the completion of a functional frequency discrimination model. Difficulties in understating the software interface made advances slow and tedious. A hard disk failure near the end of the project practically ended work on the model as much of the remaining time was spent recovering lost data and rebuilding the software interface.

IV. Discussion

Creating the speech-processing model proved to be very beneficial. The experiences gained from troubleshooting the model made it much easier to understand and alter the processing models used in the melody recognition experiments. In addition the literature review required to complete this design provided the knowledge base needed to understand the signal processing techniques that underlie each of the acoustic models.

The results for Experiment I were comparable to those found by Kong *et al.* (2004) in a similar experiment on melody identification. One difference was the percent correct for 2F processed melodies with no rhythm. The subjects in the Kong *et al.* (2004) experiment performed at near chance levels (8.33%) for this condition. The minimum performance in this experiment for the same condition was 39% correct with an average of 57%, both significantly greater than chance. For this study, the participants were chosen from within the research lab. Each subject has had experience with this type of psychophysical testing and several have design acoustic models. This abnormally high level of experience may have affected performance in the most difficult hearing tasks. In addition, because the order of testing was not random, over-training my have affected results for the final groups of testing. The results still suggest that as spectral information is decreased, listeners rely more heavily on temporal cues to identify the melodies and as Kong et al. (2004) concludes, increased spectral information results in increased melody recognition. Even for the 6/20F model where 20 channels are used but only six are stimulated, the increase in spectral range led to increased melody perception.

In Experiment II, the frequency range for the melodic stimuli was 207-523Hz. Only the low frequency pitch reversal models (PGL and PRL) affected this frequency range. Therefore it is understandable that the high and mid frequency reversal models had no significant effect on recognition scores. It is surprising however that neither low frequency model caused significant changes in recognition when compared to the unaltered 6/20 model. This may be related to the way in which the reversal frequencies where chosen for this model. The reassignments put the reversal frequencies equidistant between unaltered frequencies to maximize discrimination between frequencies and minimize changes to the pitch range. Because the first three channels of the PRL model encompass the entire frequency range of the melodic stimuli, this model simply shifts the frequencies rather than reordering them. The PGL model had greater effects on recognition because it had a reduced number of effective channels and therefore further limits the spectral information.

Though incomplete, the work done on the frequency discrimination model is the foundation for the next phase of the project. The understanding required to complete the frequency discrimination model will be essential in the creation of a success melody recognition test for the CRI. A completed stimulation program for melody recognition would enable the testing of cochlear implant patients and allow this study to be expanded.

V. Future Work

In order to increase melody perception, more spectral information must be presented to the surviving auditory neurons. One effective way to do this is to increase the number of functional channels. Yet, multichannel cochlear implants can achieve maximum effectiveness only if each of the electrodes can independently stimulate distinct populations of neurons (Throckmorton and Collins, 1998). Research suggests however, that the close proximity of the 22 channels in the Nucleus 22 implant design already creates problems with overlapping fields caused by current spread (Pfignst *et al.*, 1999*)*. Thus with current electrode and stimulation technology it would be impractical to add more channels. Although pitch reversals may not be as detrimental to melody perception as they were shown to be with speech, it is be important to do further studies on the effects of other types of channel interactions on melody perception. These studies should include more complex stimuli with broader frequency ranges.

VI. Acknowledgments

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Appendix

Figure AI: Note Frequencies for Melodies

Melody	**Notes (Hz)**													
1	292 D4	292 D4	292 D4	220 A3	245 B3	245 B3	220 A3	369 F#4	369 F#4	329 E4	329 E4	292 D4		
2	261 C4	261 C4	392 G4	392 G4	440 A4	440 A4	392 G4	348 F4	348 F4	329 E4	329 E4	292 D4	292 D4	261 C4
3	392 G4	440 A4	392 G4	348 F4	329 E4	348 F4	392 G4	292 D4	329 E4	348 F4	329 E4	348 F4	392 G4	
4	329 E4	292 D4	261 C4	292 D4	329 E4	329 E4	329 E4	292 D4	292 D4	292 D4	329 E4	392 G4	392 G4	
5	329 E4	276 C#4	329 E4	329 E4	276 C#4	329 E4	369 F#4	329 E4	292 D4	276 C#4	245 B3	276 C#4	292 D4	
6	292 D4	292 D4	329 E4	369 F#4	292 D4	369 F#4	329 E4	220 A3	292 D4	292 D4	329 E4	369 F#4	292 D4	276 C#4
7	261 C4	292 D4	348 F4	348 F4	348 F4	348 F4	292 D4	261 C4	220 A3	261 C4	348 F4	348 F4	348 F4	348 F4
8	261 C4	261 C4	292 D4	261 C4	348 F4	329 E4	261 C4	261 C4	292 D4	261 C4	392 G4	348 F4		
9	220 A3	220 A3	261 C4	220 A3	220 A3	261 C4	220 A3	261 C4	348 F4	329 E4	292 D4	292 D4	261 C4	
10	220 A3	440 A4	369 F#4	329 E4	276 C#4	329 E4	245 B3	220 A3	440 A4	369 F#4	329 E4	276 C#4	329 E4	
11	207 G#3	276 C#4	261 C4	276 C#4	348 F4	309 D#4	276 C#4	309 D#4	348 F4	276 C#4	276 C#4	348 F4	414 G#4	466 A#4
12	309 D#4	261 C4	207 G#3	261 C4	309 D#4	414 G#4	523 C5	466 A#4	414 G#4	261 C4	292 D4	309 D#4		

1. Old Macdonald had a Farm
2. Twinkle, Twinkle, Little Star
3. London Bridge is Falling Down
4. Mary Had a Little a Lamb
5. this Old Man
6. Yankee Doodle
7. She’ll be Coming ‘Round the Mountain
8. Happy Birthday
9. Lullaby, and Goodnight
10. Take Me Out to the Ball Game
11. Auld Lang Syne
12. Star Spangled Banner

Figure A2: Carrier Frequencies for Pitch Reversal Model

Filter Number	**PRL**	**PRM**	**PRH**
1	1350*	297	297
2	1768*	442	442
3	2323*	644	644
4	2031*	845	845
5	1550*	1045	1045
6	1247	2680*	1247
7	1447	3571*	1447
8	1655	4903*	1655
9	1895	4184*	1895
10	2172	3079*	2172
11	2495	2495	5744*
12	2873	2873	7884*
13	3316	3316	10823*
14	3865	3865	9238*
15	4530	4530	6730*
16	5307	5307	5307
17	6218	6218	6218
18	7285	7285	7285
19	8535	8535	8535
20	10000	10000	10000

* Frequency was reassigned to account for pitch reversals

Figure A3: Subject Information

Subject	**Age**	**Musical Experience**
S1	24	5 years band
S2	21	None
S3	22	None
S4	23	None
S5	31	Band
S6	31	None